REMARKS

Reconsideration of the present application, as amended, is respectfully requested.

By means of the present amendment, the specification, abstract and claims have been amended to place them in better form, such as correcting certain informalities in the specification, limiting the abstract to a single paragraph, and changing "characterized in that" to --wherein-- in the claims.

In the Office Action, claims 1-3, 9-11, 13, 15, 16 and 21 were rejected under 35 U.S.C. §102(e) as being anticipated by U.S. 6,078,879 (taori). Further, claims 4, 5, 12, 14, 16, 18 and 22 were rejected under 35 U.S.C. §103(a) as being unpatentable over taori in view of U.S. 5,647,005 (Wang). In addition, claims 6-8 were rejected under 35 U.S.C. §103(a) as being unpatentable over taori in view of Wang and further in view of Sluijter ("a Time Warper for Speech Signals," Proceedings of IEEE Workshop on Speech Coding Proceedings. Model, Coders, and Error Criteria, Porvoo, Finland, 20-23, June 1999, pages 150-152). Applicants respectfully traverse these rejections and submit that claims 1-26 are patentable over Taori, Wang and Sluijter for at least the following reasons.

Taori discloses a harmonic speech encoder where a speech

signal is represented by LPC parameters, a pitch value and a gain value. A refined pitch value is selected which results in a minimum error between a representation of the original speech signal, and a representation of the synthesized speech signal, as recited in the abstract. The synthesized speech signal includes a plurality of harmonically related sinusoidal components of the original speech signal, as recited in column 11, line 67 to column 12, line 9. Thus, Taori teaches minimizing the difference between two different signals, namely, between representations of the original signal and the synthesized speech signal.

In stark contrast, the present invention, as recited in independent claims 1, 9, 13, 15, 17, 19 and 21, require determining a signal that represents a frequency change of a periodical component of the very same audio signal over a predetermined amount of time. Such a signal is nowhere taught or suggested in Taori, let alone using such a frequency change signal to derive a reconstructed original signal, as recited in independent claims 13, 17 and 21.

Wang and Sluijter are cited to show compression/expansion of the audio signal, and selection of highest peak in the autocorrelation function, and do not remedy the deficiencies in Taori. Accordingly, it is respectfully submitted that independent

claims 1, 9, 13, 15, 17, 19 and 21 be allowed. In addition, as claims 2-8, 10-12, 14, 16, 18, 20 and 22-26 depend from independent claims 1, 9, 13, 15, 17, 19 and 21, applicants respectfully request that claims 2-8, 10-12, 14, 16, 18, 20 and 22-26 also be allowed over the prior art of record.

Claims 2, 10, 13 and 25 also include patentable subject matter, since Taori, Wang and Sluijter do not teach or suggest a further signal representing the frequency change which is transmitted to the receiver, where a decoder in the receiver derives the reconstructed audio signal using the frequency change included in the further signal. Column 3, lines 6-7, 12-16 of Taori merely teach a receiver having a decoder which converts the received signal to a reconstructed speech signal. Transmitting a further signal representing the <u>frequency change for use in deriving the reconstructed audio signal</u> is nowhere taught or suggested in Taori.

Rather than using a further signal representing the frequency change, Toari teaches a receiver that transmits <u>LPC codes, gain and refined pitch</u> which are used by synthesizers 94, 96 (FIG 7) <u>to synthesize</u> voiced and unvoiced speech signals, as recited on column 9, lines 12-22.

Claims 4, 12, 14, 16, 18, 20 and 22 also include patentable subject matter, since Taori, Wang and Sluijter do not teach or

suggest a time compressing the audio signal during a first part of the predetermined amount of time and for time expanding the audio signal during a second part of the predetermined amount of time, let alone teaching or suggesting doing so in such a way that the time transformed audio signal has a smaller frequency change than the audio signal.

It is respectfully submitted that Wang merely teaches that if the pitch is increased (or decreased) by increasing (or decreasing) the frequency, then the playing time is shortened (or lengthened). This is different from the recitation of Claim 4, 12, 14, 16, 18, 20 and 22, namely, compressing the audio signal during a first part of the predetermined amount of time, and for time expanding the audio signal during a second part of the predetermined amount of time.

Assuming this Wang teaching of changing pitch/frequency/time is equivalent to compressing/expanding the audio signal, Wang still does not compressing the audio signal <u>during a first part of the predetermined amount of time</u>, and for time expanding the audio signal <u>during a second part of the predetermined amount of time</u>.

Further, there is no teaching or suggestion in Wang of compressing/expanding in such a way that the time transformed audio signal has a smaller frequency change than the audio signal.

In view of the above, it is respectfully submitted that the

present application is in condition for allowance, and a Notice of Allowance is earnestly solicited.

If any informalities remain, the Examiner is requested to telephone the undersigned in order to expedite allowance.

Please charge any fee deficiencies and credit any overpayments to Deposit Account No. 14-1270.

Respectfully submitted,

By [signature]
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July 2, 2003

Enclosure: Substitute Abstract
Marked Up Abstract (Appendix A)

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Appendix A

Version with Markings
to Show Changes Made to the Abstract

~~In several types of audio coding a frequency of one or more periodical components is determined and used in the encoding process.~~

~~A first example is a coding method in which the~~ An analyzer determines frequency and amplitudes of an audio signal ~~is~~ represented by ~~a plurality of~~ sinusoids ~~from which the frequency and the amplitudes are determined by means of an analyzer (10). These amplitudes and frequencies are then transmitted~~ for transmission to ~~the~~ a receiver decoder ~~(20)~~ which ~~comprises~~ includes a synthesizer ~~(24) which~~ to reconstructs the audio signal ~~on basis of the amplitudes and frequencies of determined by the analyzer (10)~~.

~~A second example is a pitch based audio coding method, which is particularly suited for encoding speech signals. In such a pitch based encoding system, the pitch is determined in a~~ A pitch detector ~~(8) and transmitted~~ determines the pitch for transmission to the receiver ~~(16). Besides the pitch, also~~ along with the structure of the spectrum of the speech signal ~~is transmitted to~~

~~the receiver (16)~~. The structure of the spectrum is often transmitted in the form of LPC parameters~~, such as LAR's (Log Area Ratios) or LSP's (Line Spectral Pairs).~~

~~In practical audio signals, the frequency of the periodical component to be determined is not always constant, but may slightly vary over an analysis interval.~~ To correct for ~~said~~ frequency changes of the periodic component of an audio signal, ~~the system according to the invention comprises~~ a frequency change determiner~~ing means (8) which~~ determines a change of the frequency of the periodical component over the analysis period. This change of frequency ~~can be~~ is transmitted to the decoder for increasing the accuracy of the reconstruction of the audio signal. ~~Also it is possible that~~ Further, the frequency change is only used to obtain a more accurate value of the pitch.

~~Preferably the~~The frequency change is determined by using a time warper ~~(6)~~ which performs a time transformation such that a time transformed audio signal is obtained with a minimum frequency change.

~~Fig. 1~~